What is claimed is:

1. A frequency domain interpolative CODEC system for low bit rate coding of speech, comprising:

a linear prediction (LP) front end adapted to process an input signal providing LP parameters which are quantized and encoded over predetermined intervals and used to compute a LP residual signal;

an open loop pitch estimator adapted to process said LP residual signal, a pitch quantizer, and a pitch interpolator and provide a pitch contour within the predetermined intervals; and

a signal processor responsive to said LP residual signal and the pitch contour and adapted to perform the following:

provide a voicing measure, said voicing measure characterizing a degree of voicing of said input speech signal and is derived from several input parameters that are correlated to degrees of periodicity of the signal over the predetermined intervals;

extract a prototype waveform (PW) from the LP residual and the open loop pitch contour for a number of equal sub-intervals within the predetermined intervals;

normalize the PW by said PW's gain; and

directly quantize the PW in a magnitude domain without further decomposition of said PW into complex components, said direct quantization being performed by a hierarchical quantization method based on a voicing classification using fixed dimension vector quantizers (VQ's).

2. A system as recited in claim 1, wherein said predetermined interval comprises a frame.

3. A system as recited in claim 2, wherein said frame is preferably 20 ms.

4. A system as recited in claim 1, wherein said extraction of said PW sub-frame is preferably performed every 2.5 ms.

5. A system as recited in claim 1 further comprising:

a hierarchical mean-deviation vector quantization of a variable dimension PW magnitude vector is performed using fixed dimension VQs.

6. A system as recited in claim 5, wherein a seven dimensional mean vector is derived as a seven band subband average of the PW magnitude vector.

7. A system as recited in claim 6, wherein a ten dimensional deviation vector is derived as the difference between magnitude and mean vectors for pitch harmonics between the range of about 1 to 10.

8. A system as recited in claim 7, wherein said hierarchial mean-deviation vector quantization comprises switched backward prediction of the variable dimension PW magnitude vector using fixed dimension VQs.

9. A system as recited in claim 8, wherein said hierarchial mean-deviation vector quantization comprising switched backward prediction involves the following:

a prediction mode for mean and deviation selection based on a voice/unvoiced classification system derived from a voicing measure parameter;

a predictive VQ of the mean is performed once per frame in the voiced mode;

a predictive VQ of the deviation is performed twice per frame in the voiced mode;

a nonpredictive VQ of the mean is performed twice per frame in the unvoiced mode; and

a non predictive multistage VQ of the deviation is performed twice per frame in the unvoiced mode.

10. A system as recited in claim 9, wherein said signal processor is adapted to perform reconstruction of the PW magnitude vector at a decoder by adding said PW mean and deviation components.

11. A system as recited in claim 10, wherein said signal processor is adapted to linearly interpolate between said PW magnitude vectors in order to reconstruct said PW magnitude vector at intermediate subframes.